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**APPLICATION FOR LETTERS PATENT**

**TITLE: HIGH BANDWIDTH REAL-TIME OSCILLOSCOPE**

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## **HIGH BANDWIDTH REAL TIME OSCILLOSCOPE**

### **Cross-Reference To Related Applications**

This application claims the benefit of U.S. Provisional Patent Application 60/420,937 filed October 24, 2002, the entire contents of which are incorporated herein  
5 by reference.

### **Field of the Invention**

The present invention relates to a high bandwidth real-time digital sampling oscilloscope (DSO) incorporating mixing (or heterodyning) to increase the bandwidth of a typical oscilloscope design with limited bandwidth.

### **Background of the Invention**

A digital sampling oscilloscope (DSO) is the primary tool utilized by engineers to view signals in electronic circuitry. As signals get ever faster, it is very beneficial to have DSOs capable of digitizing, displaying and analyzing these faster signals. The capability of a DSO to digitize fast signals is determined by its bandwidth and sample rate. The  
15 sample rate is the number of samples points taken of a waveform in a given amount of time and is inversely proportional to the sample period – the time between samples. If a sinusoidal frequency sweep is performed from DC up to higher frequencies, the bandwidth is defined as the frequency at which the signal displayed on the DSO screen is approximately 30% smaller than the input sine-wave.

20 Since one of the uses of the DSO is to design and analyze new electronic devices, high end DSOs must operate at speeds much higher than the present state of the art in electronics. These speeds are generally unachievable through brute-force methods, such as simply providing ever-faster sampling chips, and many methods are employed to overcome this situation. One of the most common methods is a method inherent in the  
25 design of the original oscilloscope – that of triggering repeatedly on a periodic event. If an event is frequently, periodically repeating, the waveform at the time of the event can be repeatedly displayed on the screen. Furthermore, data from multiple trigger events average together to provide a good view of the waveform. This technique is the

underlying method of a conventional sampling scope. A sampling scope repeatedly triggers on an event and acquires only a few points of the waveform (sometimes only one point of the waveform) on each trigger event. After repeated triggers, the points are reassembled according to the sampling algorithm to form a very high “effective” sample rate version of the waveform. Relatively low sample rates are utilized for each trigger event, and very high bandwidth samples may be generated. Furthermore, the repeated trigger events enable averaging, which can be utilized to increase the signal-to-noise ratio (SNR) and therefore enable further bandwidth increases. However, such a sampling scope presupposes a repetitive input signal so that the representation of the waveform can be generated over many triggers.

A common problem in complex signal analysis is that a signal that is to be analyzed is often not repetitive. In fact, it is very often the case that a non-repetitive event is the cause of some failure in an electronic system. It is the function of the test equipment to help the user identify the cause of the failure. Therefore, a piece of test equipment that requires repetitive signals is of limited usefulness. For example, sometimes the trigger event happens only once, such as in the analysis of bomb blasts. Frequently, however, the trigger event happens repeatedly, but the signal around the trigger event is different. Situations like this require a DSO capable of high bandwidth and sample rate with only a single trigger event. A DSO with these characteristics is called a real-time scope, and acquisitions taken utilizing only a single trigger event are called single-shot acquisitions. The distinction between the sampling scope and the real-time scope is an important one because the tricks that can be utilized to digitize a repetitive waveform are not available to the real-time DSO designer. In general, a real-time DSO is more useful because it does not require the input signal to be repetitive. However, the primary limitation is that the bandwidth of the real-time scope is limited.

In real-time DSO design, the method in common use for overcoming sample rate limitations is the method of interleaving. This method utilizes multiple digitizing elements that sample the same waveform at different points in time such that the waveform resulting from combining the waveforms acquired on these multiple digitizers forms a high sample rate acquisition. Most high-end real-time DSOs have very high

sample rates achieved through the use of interleaving and most are capable of “oversampling” an input waveform.

Oversampling is defined as sampling a waveform at a rate whereby virtually no amount of signal content is present at a frequency above one half the sample rate. For example, a DSO with a bandwidth of 6 GHz that does not allow any signal in with a frequency at or above 10 GHz would be sufficiently sampling the waveform at a sample rate of 20 GHz. Any sampling of the waveform above this sample rate would result in an oversampled waveform. Oversampling is not inherently bad, just unnecessary because much more elegant methods can be utilized to produce the highly sampled waveform.

The criteria for sufficiency of sample rate, outlined by Nyquist, states that if a waveform is sampled at a sufficient rate, then the exact analog waveform can be reconstructed. In other words, once the waveform is sufficiently sampled, the waveform can be reconstructed as if physically digitized at any sample rate.

Generally, in real-time DSOs, the interleaving is controlled through a method called channel combination. Combining channels means that the digitizing resources of multiple channels are utilized together to digitize a single waveform. Most often, channel combination is utilized to interleave multiple digitizers for the purpose of increasing the sample rate, but as the acquisition memory is generally connected to individual digitizers, this method is also utilized sometimes to increase the length of the acquisition.

While techniques are generally available for designing high sample rate systems, bandwidth is another issue. Bandwidth is typically dealt with through direct application of very high-speed electronics. In situations where electronics are simply not fast enough, usually the attainment of high bandwidth is achieved by making tradeoffs that the customer simply must accept. For example, higher bandwidth is achievable by removing protection circuitry at the front-end of the scope, thereby making it more susceptible to damage from static discharge or signal overdrive. Also, limitations are placed on the user regarding the range of input signals (to allow for attenuators and active gain components to be eliminated). The tradeoffs foisted on the scope user are often unpalatable, but endured reluctantly by the user with high-bandwidth requirements.

Despite this situation, the fact remains that attempts made to reach high bandwidths are often done at the expense of the overall usability of the scope. In other

words, a high bandwidth scope can often not be utilized in a general-purpose manner. Finally, the fact remains that even with every possible trade-off, the bandwidth needs of the real-time scope user are sometimes so high as to be unachievable with the current state of the art.

5 **Summary of the Invention**

This invention pertains generally to systems that digitize waveforms; and more specifically systems that convert an analog input signal to a digital output signal whereby the digital signal consists of an array of numbers that represent the amplitude of the analog waveform at known times. This invention also pertains to systems with limited bandwidth where there is a need for higher bandwidth. This invention therefore addresses systems incapable of accurately digitizing very rapidly changing signals. The most specific application of this invention is to the high-end real-time DSO where extremely high demands are placed on the speed (and bandwidth) of signals digitized in a single-shot acquisition.

15 It is an object of the invention to demonstrate a method and apparatus whereby the bandwidth of a digitizing system can be increased.

As mentioned previously, channels are often combined within a DSO for the purpose of increasing the sample rate and acquisition memory length. This invention puts forth a method of utilizing channel combination for the purpose of increasing bandwidth.

20 Still other objects and advantages of the invention will in part be obvious and will in part be apparent from the specification and the drawings.

The invention accordingly comprises the several steps and the relation of one or more of such steps with respect to each of the others, and the apparatus embodying features of construction, combination(s) of elements and arrangement of parts that are adapted to effect such steps, all as exemplified in the following detailed disclosure, and the scope of the invention will be indicated in the claims.

**Brief Description Of The Drawings**

For a more complete understanding of the invention, reference is made to the following description and accompanying drawings, in which:

Figure 1 is a block diagram depicting a digitizing system constructed in accordance with the invention;

Figure 2 is a block diagram showing one possible method of the extension of this technique to 4 channels using a mixing frequency that is at the low side of the frequency band of interest (low side conversion); and

Figure 3 is a block diagram showing another possible method of the extension of this technique to 4 channels using a mixing frequency that is at the high side of the frequency band of interest (high side conversion).

### **Detailed Description Of The Preferred Embodiments**

Figure 1 is a block diagram showing a high bandwidth digital oscilloscope architecture according to the present invention. Figure 1 shows two channels of a DSO combined to digitize waveforms in a manner that effectively doubles the system bandwidth. It should be understood that the bandwidth can be tripled, quadrupled etc. by utilizing three, four or more channels in combination.

An input signal 100 is provided at the input. Viewed from the frequency-domain perspective, the input signal might have a frequency content shown as 103. In a standard configuration input signal 100 directly enters a first channel, CH1 at 104. This signal passes through an analog front end 105, and on to an ADC 106 which digitizes the waveform. The channel has a finite bandwidth, as shown by 107 which results in a digitized waveform of finite bandwidth 108. For the purpose of future explanation, the cutoff frequency at which the bandwidth is limited is designated as F. In a conventional digitizer the CH1 channel (104) and a CH2 channel (109) are combined utilizing preferably an additional output of the CH1 channel (104) front-end (105) connected to an additional input of a CH2 channel ADC (111- connection not shown) for the purpose of doubling the sample-rate and memory length of the acquisition. A less preferable connection combines the channels through a 50 Ohm power splitter at the two channel inputs 104 and 109. Neither of these methods used in the current state of the art has will increase the bandwidth. If there is any effect at all, it is to decrease the bandwidth somewhat. This is a side effect and not generally desired. The effect can be minimized with careful design.

The present invention involves the addition of additional analog circuitry 102 between the input 101 and the two channels 104 and 109 and downstream processing of the digital data to account for this additional hardware. In accordance with the present invention, the signal at the input 101 with example frequency content 103 enters a 50  
5 Ohm power splitter 112. The splitter 112 provides the 50 Ohm termination to the input signal and provides at its two outputs, the same signal attenuated by 6 dB. One output of splitter 112 directly connects to CH1 channel (104). The signal proceeds through front-end 105 and is digitized in the normal manner by ADC 106. Since the combination of front-end 105 and ADC 106 is bandwidth limited, as shown in 107, this results in a  
10 bandwidth limited acquisition with frequency content shown by 108. CH1 channel (104) is designated as containing the low frequency (LF) portion of the signal. The path through the other output of splitter will be described below and is designated as the high frequency (HF) path.

The addition of the power splitter is the only additional component directly in the  
15 signal path of CH1 channel (104). In other words, the low frequency signal path with the splitter removed looks identical to the signal path of the DSO not utilizing this invention. Very high quality splitters with very high bandwidth are readily commercially available and as such, do not serve to degrade the signal path, except for the fact that they decrease the signal strength by 6 dB.

The HF signal path will now be explained in greater detail. It is well known that  
20 frequencies can be shifted through the use of a process called mixing or heterodyning. This process is called “frequency translation”. Mixing is achieved through the time-domain multiplication of a signal with another sinusoidal signal. It is well known that if a sinusoid with a frequency  $f_0$  is mixed with another sinusoid with a frequency  $f_1$ , the  
25 result is two sinusoids at sum and difference frequencies (i.e. sinusoids at frequencies  $f_0+f_1$  and  $f_0-f_1$ ) with each sinusoid being half the amplitude of the product of the amplitudes of  $f_0$  and  $f_1$ :

$$[A_0 \cdot \cos(2 \cdot \pi \cdot f_0)] \cdot [A_1 \cdot \cos(2 \cdot \pi \cdot f_1)] \rightarrow \frac{A_0 \cdot A_1}{2} \cdot [\cos(2 \cdot \pi \cdot (f_0 + f_1)) + \cos(2 \cdot \pi \cdot (f_0 - f_1))]$$

In order to utilize these principles, the signal from the second output of splitter 112 enters a high pass filter 113. High-pass filter 113 is designed to reject to the greatest extent possible all frequencies below frequency  $F$  as shown in 114. The result of high pass filtering the input signal with frequency content as shown in 103 is shown in 115. The output of 113 is mixed with a sinusoid at frequency  $F$  utilizing mixer 116. The result of mixing the signal with frequency content shown by 115 with the mixing frequency  $F$  shown in 117 is the frequency content shown in 118. 118 shows that two images of the content shown in 115 are produced at the sum and difference frequencies, as noted above. In cases where the cutoff of the high-pass filter is inadequate, the mixing frequency could be chosen slightly higher such that a dead-band is utilized to prevent the low frequency edges of the high-pass filter output from folding back into the pass-band. Any increase in mixing frequency, while providing margin, will also serve to degrade the maximum bandwidth achievable.

The output of mixer 116 connects to CH2 channel (109) – the high frequency channel. The signal passes through a front-end 110 and is digitized by an ADC 111. Since like CH1 channel (104), the combination of front-end 110 and ADC 111 is bandwidth limited, as shown in 119 this results in a bandwidth limited acquisition with frequency content shown by 120.

The acquisition through ADCs 106 and 111 occur simultaneously in parallel, so channels CH1 and CH2 are simultaneously acquired.

To summarize, the signals seen at the inputs to each of the channels are as follows. The LF CH1 channel (104) sees the input waveform directly. The HF CH2 channel (109), sees only the frequency content in the frequency band extending upward from  $F$ . Through the use of heterodyning, it sees the frequency content of the input signal at  $F + \Delta F$  at the frequency location  $\Delta F$ . Thus, the input signal has been *mixed down*. Although there were two images (seen in 118), the second image (and part of the first image) were rejected due to the finite channel bandwidth 119. Said differently, the LF CH1 channel acquires the low frequency content of the input signal from  $0$ - $F$ , while the HF CH2 channel acquires the frequency content of the input signal from  $F$ - $2 \cdot F$ . This signal is mixed down from frequency band  $F \rightarrow 2 \cdot F$  to the range of  $0 \rightarrow F$  so it “fits”



into the bandwidth of the front end. It can be seen, twice the frequency content of the signal has been made to “fit” into the bandwidth of the scope.

Both the LF and HF signals are digitized by the scope in the normal manner. It is assumed that both channels are sufficiently sampling with respect to the channel  
5 bandwidth. In other words, each channel is sampling at a sample rate ( $F_s$ ) such that virtually no frequency content can get through the channels above  $F_s/2$ . This might be accomplished through the use of internally interleaving channel digitizers or through the combination of other channels, as mentioned previously. Since a sufficiently sampled channel allows for the complete reconstruction of the signal at any sample rate, each  
10 channel is upsampled to a sample rate that at least sufficiently samples the resulting acquisition utilizing upsamplers 121 and 122. In the case where a channel is just sufficiently sampling based on the frequency response of the channel, the acquisition on each channel is upsampled by a factor of 2 when two channels are combined, since the system bandwidth will be effectively doubled in the end. This upsampling is performed  
15 utilizing a method such as SinX/X interpolation to interpolate every other sample point. The method and validity of this method of interpolation is well known to those skilled in the art.

The data from the HF CH2 channel is mixed at a mixer 123 digitally (i.e. numerically using a software program and floating point arithmetic) with a sinusoid with  
20 the same frequency  $F$  (124) as utilized by analog mixer 116 in the analog HF signal path. The result of mixer 123 is two images of the HF signal shown by 125. Each image appears at frequencies from 0- $F$  and from  $F-2\cdot F$ . The first, low frequency image is mirrored about  $F$  and is unusable. The second image is a replica of the actual high frequency content of the input signal. The output of digital mixer 123 passes through an  
25 image reject filter 126, which has a frequency response shown by 27. The result is the frequency content shown in 128.

The result at this point is two digital waveforms, one representing the low frequency portion of the input signal 108 and the other representing the high frequency portion of the input signal 128.

30 Since both signals passed through an imperfect channel, they are equalized separately to compensate for non-ideal magnitude and phase characteristics of the front-

end and digitizing systems. The equalizer for the CH1 channel (129) is shown with its response 130 being an ideal low-pass filter. This results in no change between the frequency content shown in 108 and the equalized content shown in 131 with the understanding that this would not necessarily be the case if there were imperfections in the signal 131. The equalizer for the CH2 channel is shown integrated with the image reject filter 126. Both equalizers also have the difficult job of preparing the signals to accommodate the cross-over from LF to HF.

Finally, the waveform resulting from the LF channel equalizer 129 is added to the waveform resulting from the HF channel equalizer 126 by the summer 132. The result of this addition is shown by adding the LF frequency content in 131 to the HF content in 128, shown graphically in 133. This forms a high-bandwidth, high-sample rate acquisition at the output 135 with the frequency content as shown in 134.

A detailed numerical example stepping through this process is provided in Appendix A.

To summarize the effect, the input signal with frequency content 103 normally would be digitized by one channel to form an acquisition with frequency content shown in 108. Instead, as a result of this invention, two channels were utilized and the resulting acquisition as twice the bandwidth as demonstrated by the signal frequency content shown in 134. Note that the two other benefits of interleaving – that of doubling the sample rate and memory length are still achieved by this invention.

Multiple channels could be combined in similar arrangements whereby the system bandwidth is increased by a factor equal to the number of combined channels. An example showing how this technique may be used to extend the bandwidth four times using four channels is shown in Figure 2. Figure 2 depicts a low side conversion. In this case, each frequency band in 201 (A, B, C & D) is translated down to be digitized by channels 1, 2, 3 and 4 respectively. The filter blocks shown (202) are optional. The goal is to sufficiently isolate the desired frequency band. This can be done, in this example, using a highpass filter, a bandpass filter or no filter at all. In the case of no filter, there will be “images” of the adjacent frequency band that will be digitized by the channel. These can be removed using DSP techniques in the recombination DSP block (203).

This recombination technique is shown for a low side conversion (using three channels) in Appendix A.

Another example of how this technique can be applied is shown in Figure 3. Figure 3 depicts a high side conversion. Careful inspection of Figure 3 will show that the major difference between Figure 2 and Figure 3 is the frequency used to “translate” the frequency band of interest into the frequency band of the acquisition channel. Figure 2 uses a frequency on the low side of the band of interest ( $F_1$  to translate frequency band “B” to the frequency band of the acquisition system) and Figure 3 uses a frequency on the high side of the band of interest ( $F_2$  to translate frequency band “B” to the frequency band of the acquisition system). Note that the translated frequency band is “reversed” if high side conversion is used. That is, the highest frequency in frequency band “B” ( $F_2$ ) becomes the lowest frequency in the translated band. This will be corrected in the reconstruction by using  $F_2$  as the up-conversion frequency in the DSP reconstruction of the signal. This recombination technique is shown for a high side conversion (using two channels) in Appendix B.

In either case shown in Figs. 2 and 3, it is required that the phase of the translation frequency is known in order to reconstruct the original signal. This can be accomplished, by example, by summing a pilot tone into the signal channel, or locking the mixer phase to the sample clock.

It should be obvious to one skilled in the art that there are many combinations of translation frequencies and filter choices that will accomplish the objective of this invention. Each has different tradeoffs and implementation considerations depending on the specific application.

It will be understood that the above description of the present invention is susceptible to various modifications, changes and adaptations, and the same are intended to be comprehended within the meaning and range of equivalents of the appended claims. The most obvious modification, for example, is the use of more than two channels.

It will thus be seen that the objects set forth above, among those made apparent from the preceding description, are efficiently attained and, since certain changes may be made in carrying out the above method (process) without departing from the spirit and

scope of the invention, it is intended that all matter contained in the above description shall be interpreted as illustrative and not in a limiting sense.

It is also to be understood that the following claims are intended to cover all of the generic and specific features of the invention herein described and all statements of the  
5 scope of the invention which, as a matter of language, might be said to fall therebetween.